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Date: 10/31/2004

Description of Documents Translated:

Job Number T15018: EP patent 0 600 164 A1 (Method for the Improvement of Transmission Properties of an Electroacoustic Device)

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(19) European Patent Office

(11) Publication number: **0 600 164 A1**

(12) **EUROPEAN PATENT APPLICATION**

(21) Application Number: **93113875.4**

(51) Int. Cl.^{5}: **H04R 3/02**

(22) Date of Filing: **8/31/93**

(30) Priority: **9/8/92 DE 4229912**

(43) Date of Publication of Application:
6/8/94 Patentblatt 94/23

(84) Designated Contracting States:
AT BE CH DE ES FR GB IT LI NL SE

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(54) **Method for the Improvement of Transmission Properties of an Electroacoustic Device**

(57) In a telephone with a hands-free speaking device and compander control, the characteristic curve (LU2 = f(LU1)) of the compander is adjusted in such a manner that correct operation is ensured with the usual system parameters. Control errors will occur if very high voice, reproduction, or noise levels occur. An additional control of the characteristic curve is effected with greatly differing higher levels of the said type by deriving a separate voice and noise level from the transmitted signal and a control variable determined thereupon. As a result, intelligibility is still ensured even at high levels.

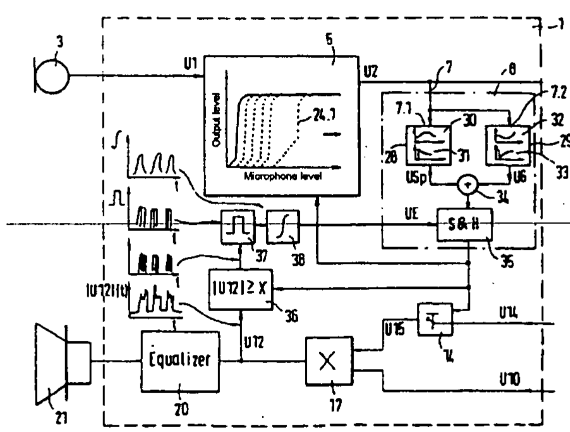


FIG. 2

The present invention relates to a method for improving the transmission properties of an electroacoustic device provided with adaptive dynamic control in accordance with the preamble of claim 1.

Such a method is known from DE A 37 24 346. The compander compresses the signals produced by the microphone, called transmitted signals hereafter, to a uniform signal level insofar as they are above a nominal value, amplifies them to the maximum insofar as they are at the nominal value, and expands them insofar as they are below the nominal value. The compander characteristic curve LU2 = f(LU1) has a steep expansion range and a flat compression range. The characteristic curve can be displaced and optionally also the steepness of the characteristic curve in the expander range can be changed by feeding the output signals back to the input of the compander through a low-pass or band-pass filter. In addition, the characteristic curve displacement can be produced in the receiving branch as a function of volume control. As a rule, the system parameters are set by the manufacturer to the normal acoustic conditions in an office environment.

If the conditions deviate therefrom, e.g., a very high noise level, a manual adjustment must be made, because an increase in ambient noise above the starting point of the expansion causes the noise to be transmitted with increased amplification.

The object is to be achieved with the present invention to improve further the known method or a device suitable therefor in such a way that both when raising the speaking volume and with increasing ambient noise, good communication is achieved without the participant having to take any special measures.

This object is achieved by means of the features set forth in claim 1.

This voice recognition derived from the transmitted signal and noise recognition, the special type of voice recognition, and the control of both the compander and the receiving volume by means of the control variables produced by voice and noise recognition achieve that the voice to be transmitted above the nominal value is always sent at a constant level, and with increasing ambient noise the receiving volume is raised but the noise level is reduced by the expander function, without any manual interventions in the process or in the device having to be made.

In fact, it is already prior in the art, e.g., from [German Unexamined Patent Application] DE-OS 24 56 468, to improve the intelligibility of an electroacoustic sound reproduction, during which ambient noises occur, to provide an additional detector that determines interfering noises, and with increasing noise level to raise the reproduction sound intensity. However, there is no reduction in the interfering noises and no production of a control component from the voice and interference signals of the transmitting branch. However, an interference signal-dependent and frequency-dependent useful signal compression occurs to diminish the difference between loud and quiet places in the reproduction.

Other advantageous details of the invention are set forth in the dependent claims and described in the following text based on the exemplary embodiments illustrated in the drawings.

The figures show the following:

Fig. 1	a block circuit diagram of a hands-free principle of a telecommunication terminal with adaptive dynamic control,
Fig. 2	an expanded block circuit diagram,
Figs. 3 and 4	each a flow diagram of the mode of operation of a compander control,
Fig. 5	the time course of the transmitted signal at the compander output,
Fig. 6	a characteristic curve field of a possible characteristic curve form,
Figs. 7 and 8	the basic configuration of a voice detector or a noise detector with the graphic representation of the filter curve and the time constant of the same,
Fig. 9	a basic circuit diagram of a telecommunication terminal with the expansion of the invention,
Fig. 10	another block circuit diagram of the input section, e.g., of a terminal, and
Fig. 11	a possible characteristic curve field for this.

In the block circuit diagram of Fig. 1, the number 1 designates an electroacoustic device in the form of a communication terminal with a hands-free device. It has a transmitting branch 2 with a microphone 3, the transmitted signal U1 of which is applied to the input 4 of a compander 5 having an amplifier, compressor, and expander. The modified transmitted signal U2 is supplied from the output 6 of the compander 5 to a transmission line. In addition, it is applied to one input 7 of a compander control 8. The transmitted signal U1 is applied to a second input 9 of the same. The output 10 of the compander control 8 is connected to the control input 11 of the

compander 5 and moreover to one input 12 of two inputs 12 and 13 of a controllable damping unit 14. A control voltage U14 which can be adjusted by means of a volume control is located at input 13.

The voltage U15 emitted at output 15 of the damping unit 14 is supplied to the one input 16 of an amplifier and multiplier 17, to the other input 18 of which an incoming signal U10 is applied. The amplified incoming signal U12 emitted by the multiplier 17 at output 19 reaches a loudspeaker 21 of the receiving branch 22 via an equalizer 20. The incoming signal U12 is supplied to a third input 23 of the compander control 8. The mode of action of the compander corresponds to the principle inferred from [German Unexamined Patent Application] DE-OS 37 24 346. When the terminal 1 is started up, if there is neither a transmitted signal U1 nor an incoming signal U12, a characteristic curve 24 of the expander is established, which has an initial range 25 of constant amplification, a steep expansion range 26 starting at point p1 and extending up to p2, a subsequent compression range 27, extending from p2 to p3, and a subsequent approximately constant compression level. This characteristic curve is shown as a solid line in Fig. 1 within the compander block 5.

If a transmitted signal U1 of sufficiently high level occurs, it is compressed to a constant volume, because the compander 5 is operating in the compression range. The emitted, compressed transmitted signal U2 is divided in the compander control 8 into a voice component and a noise component. The peak value is detected from the voice component, and a voice level that corresponds to said level and a voice level control signal are produced or calculated from it.

The noise level is determined from the noise component, and a noise level control signal corresponding to said level is produced or calculated. This is added to the voice level control signal and supplied as control signal U6 to the compander 5 for controlling the characteristic curve LU2 = f(LU1). If the control signal U6 increases, the characteristic curve in the drawing is displaced to the right, which is indicated by the broken characteristic curves. The displacement of the characteristic curves occurs according to the invention in such a way that the determined noise level is always below the starting point p1 of the expansion range 26. This occurs, e.g., by multiplying the voice and noise level by appropriately selected weighting factors.

Through the displacement of the characteristic curve to the right the maximum sensitivity is reduced—displacement of point p3 to the right—and higher noise levels are further dampened by the concurrently displaced expander range. At the same time, the control signal U6 is supplied to the damping unit 14 and the amplification in the receiving branch 22 is increased, corresponding to an increased damping in the transmitting branch 2 due to the displacement of the characteristic curve to the right. The maximum degree of amplification depends on the existing acoustic coupling between the loudspeaker 21 and microphone 3.

If there is an incoming signal U10, it is supplied via the multiplexer 17 amplified to the loudspeaker 21. At the same time, it reaches the input 23 of the compander control. It is determined here whether the receiving level is greater or less than a preset threshold value. If it is smaller, the compander control is continued and the incoming signal U12 is changed according to the control signal U6.

Is the receiving level is greater than the threshold value, the compander control is paused and the compander control is run with the already present control signal U6 until the incoming signal U12 falls below the threshold value.

As a result, a mutual effect by the incoming and transmitted signal is avoided and the intelligibility and thereby hands-free property of the terminal 1 are clearly improved.

Further improvement can occur by supplying the transmitted signal input voltage U1 to the compander control 8. As a result, abruptly strong signal changes can be rapidly compensated.

A basic circuit diagram is shown in Fig. 2 for carrying out the aforementioned method. The compander control 8 has a voice detector 28 with an input 7.1 and a noise detector 29 with an input 7.2. Both inputs 7.1 and 7.2 are supplied with the compander output voltage—transmitted signal U2.

The voice detector 28 contains a voice frequency filter, which is shown by the filter curve 30. Expediently, the voice frequency filter is designed so that it responds primarily to the strongest frequencies in the voice frequency spectrum. In addition, it is designed so that it has a very short rise time constant, for example, 1 ms to 100 ms, and a long fall time constant, for example, 200 ms to 5 s, as is illustrated in Diagram 31. This assures that substantially only voice signals are detected and a voice level control signal USp, corresponding to the voice level, is derived or calculated from the burst and the peak value of the transmitted signal.

The noise detector 29 preferably has a noise filter, the filter curve 32 of which runs at least approximately opposite to the voice frequency filter curve 30. As a result, further improved decision and separation between voice and noise signals are achieved. In addition, the noise filter is designed so that it has a long rise time constant, for example, 200 ms to 5 s, and a short fall time constant, for example, 1 ms to 100 ms, so that the response is improved to noises occurring for a usually longer time at a relatively constant level (Diagram 33 in Fig. 2). A noise level control signal UG is maintained or calculated in accordance with the determined level.

The voice level and noise level control signals USp and UG, after appropriate weighting, are supplied to an adder 34, which emits the control signal U6 to a sample-and-hold element 35. This is controlled via the threshold detector 36, a pulse former 37 downstream to said detector, and a subsequent integrating element 38 by the control signal UE derived from the incoming signal U12. A monolithic latch is used expediently as the impulse former 37 for time dilation. The sample-and-hold element 35 retains the instantaneous value of the level of the control signal U6, if the control signal U6 is generated according to the incoming level exceeding the threshold value x. As a result, the receiving level threshold value x is changed or is changeable in accordance with the control signal U6. The receiving level threshold value x is set as a function of the anti-sidetone. Figure 3 shows the flow diagram of the method by way of example. After starting, the basic adjustment of the characteristic curve LU2 = f(LU1) is made with the points p1 and p2 according to the determined values Up1 and Up2. It is then determined whether there is an incoming signal U10. If it is found, the anti-sidetone is controlled. If no incoming signal is found, it is then determined whether a burst occurs in the transmitted signal. If this does occur, the peak value is determined by the voice detector and used to calculate the point p2 of the compression range 27 and the determined or calculated value is used to generate the control voltage U6.

If no burst is found, it can be assumed that this is noise. The noise detector 29 determines the noise level. Next, it is determined whether this is higher than the value corresponding to the specific starting point p1 of the expansion range 26, and if a higher value is determined, the control value Up1 corresponding to point p1 is determined or calculated and also used to generate the control voltage U6.

The flow diagram shown in Fig. 4 shows the weighting of the voice peak value with a weighting factor k1 in the voice recognition branch, and the weighting of the value, corresponding to the noise level, with a weighting factor k2 in the noise recognition branch. Furthermore in the voice recognition and the noise recognition, the corresponding control signal value is averaged from the instantaneous value, the previously present value U6* . a, and multiplication by the factor 1 / (a + 1) and used to generate the control signal U6. This improves the continuity of the control process.

The course of the output level LU2 is shown in Fig. 5 as a function of time and the values for the points p1, therefore the start of the expansion range 26, and p2, therefore the end of the expansion range and the beginning of the compression range 27, are indicated. Figure 6 shows the possible characteristic curve field for this.

Figure 7 shows the basic design of a voice detector 28 with a band-pass filter 39 as the voice filter, the filter curve 30 shown above this, and the used low-pass filter 40 which has the function of the rise and fall time shown in diagram 31. Filter curve 30 shows that the band-pass filter 39 as the voice filter preferably transmits the strongest frequencies in the voice spectrum. The time function shown in diagram 31 can be realized by means of the circuit design, shown below, of the capacitor C, the resistor R_x in series with diode D, and the resistor R_y, whereby the series connection of R_x and D is made at the high point of the parallel connection of R_y and C.

The basic structure of a noise detector 29 is show in Fig. 8. Its band-pass filter 41 is designed so that it has the filter curve 32, which is opposite to the voice detector filter curve 30. The low-pass filter 42 on the output side is designed so that a curve 33 with a long rise and short fall time constant is obtained, which can be realized by means of the circuit design, shown below with the parallel branch connected at the high point of the capacitor C from the series connection of a resistor R_x with a diode D and the resistor R_y parallel thereto.

The basic circuit diagram in Fig. 9 shows the structure of a compander 5 with a compressor element 43 and an expander element 44 and a control voltage U13, derived from the incoming signal U12, for controlling the anti-sidetone. The compander control of the invention occurs by means of the additional use of the voice and noise detectors 28, 29, already described in Fig. 2, the sample-and-hold element 35, and the threshold detector 36 with pulse former 37 and integrator 38.

In the exemplary embodiments described thus far, the branching of the transmitted signal to the voice and noise detector 28, 29 occurs at the output 6 of the compander 5. The branching, however, can also occur according to Fig. 10 between microphone 3 and the input 4 of the compander 5.

Instead of the analog components, digital components can be used and the level converted to digital signals and the values, to be determined, for characteristic curve displacement are calculated. For this purpose, a digital signal processor can be used, in the allocated memory of which the characteristic curve form is saved as digital words, e.g., in 16 bit width, etc. In Fig. 10 y(k) denotes the sampled values of the microphone signal U1, whereby k are the sampling time points. $\bar{y}_s$ is the digital control value, formed from the short-time average of the transmitted signal U1 or U2, for calculating the amplification factor. It can be calculated using the following formulas, where a_r denotes the coefficient for the time constant for rising levels and a_f for falling levels; specifically, for rising levels:

$$\bar{y}_s(k) = (1 - a_r)\,|y(k)| + a_r\,\bar{y}_s(k - 1),$$

if the amount of the instantaneous transmitted signal $|y(k)|$ is greater than the previous value $y_s(k - 1)$, and for falling levels:

$$\bar{y}_s(k) = (1 - a_f)\,|y(k)| + a_f\,\bar{y}_s(k - 1),$$

if the amount of the instantaneous transmitted signal $|y(k)|$ is less than or the same as the previously calculated value, $\bar{y}_s(k - 1)$.

These formulas describe in principle a first-order recursive filter. They describe a nonlinear filter through the relationship between the amount of the transmitted signal $|y(k)|$ and the previously determined control signal value $y_s(k - 1)$, which serves as the criterion for rising and falling levels and determines which parameter a_i is used. These parameters a_i determine the time constant of the estimation process and are selected so that the rise time of the slope of the input level is about 5 to 20 times smaller than the falling slope.

This estimated value $\bar{y}_s(k)$ forms the basis for calculating the amplification factor $g(\bar{y}_s)$. This calculation, however, is still dependent on the position of the starting point p2 of the compression, which is then designated as the threshold value y_o. Furthermore, the steepness of the expansion range is determined by a factor p and the expansion range by the factor w, from which the expansion swing results. Based on these factors the individual range 25, 26, 27 of the characteristic curve can be described by the following formulas:

a) range 25 up to starting point p1 of the expansion:

$$g(\bar{y}_s) = C_1,$$

if the control signal $\bar{y}_s \leq wy_o$

b) expansion range 26 of p1 to p2 or y_o:

$$g(\bar{y}_s) = C_2\,\bar{y}_s^{\,p-1},$$

if $wy_o < \bar{y}_s \leq \bar{y}_o$, and

c) compression starting at p3 or y_o:

$$g(\bar{y}_s) = C_3 / \bar{y}_s,$$

if $y_o < \bar{y}_s$.

In the exemplary embodiment according to Fig. 10, therefore the calculation of the short-time value average $\bar{y}_s$ of the transmitted signal y(k) is calculated according to the indicated formulas. This value of $\bar{y}_s$ is compared with the present threshold value yo. If it smaller than said value, it is concluded that this is not voice signal. Accordingly, in this case, the long-time term of the noise portion $\bar{y}_{ln}$, which is multiplied by the weighting factor $w_n(k)$ and is kept available for evaluation, is calculated.

If the average of the short-time term $\bar{y}_s$ is greater than the threshold value y_o, then the average of the long-time term of the voice portion $\bar{y}_{ls}$ is calculated, if necessary multiplied by a weighting factor $w_s(k)$, and added to the long-time term $\bar{y}_{ln}$ of the noise portion. The calculated value is the new nominal value for calculating the threshold value y_o. This is compared with a calculated average derived from the incoming signal x(k) of the short-time term $\bar{X}_s$ of the incoming signal x(k), multiplied by a loudspeaker-microphone coupling factor C_{lm}. If the calculated value for the nominal value of y_o is greater, then the same is calculated. The obtained value y_o is used together with the calculated value $\bar{y}_s$ for calculating the amplification factor $g(\bar{y}_s)$ and the new characteristic curve is then calculated.

Figure 11 shows this correlation between the incoming-transmitted signal $\bar{y}_s$ (U1 in Figs. 1, 2, and 9) and the amplification $g(\bar{y}_s)$, which corresponds to the relationship between the input and output signal of the transmitting branch. The threshold value y_o or the point p2 is to be selected so that the noise level does not reach the expansion range 26. An adaptation to the ambient noise must therefore still occur. To accomplish this, the average values of the

long-time terms of the voice signal, $\bar{y}_{ls}$, and the noise signal, $\bar{y}_{ln}$, are necessary, which can be calculated using the following formulas:

$$\bar{y}_{ln}(k) = (1 - a_l)\,|y(k)| + a_l\,\bar{y}_{ln}(k - 1),$$

if $\bar{y}_s(k) \leq y_o(k)$,

$$\bar{y}_{ln}(k) = \bar{y}_{ln}(k - 1),$$

if $\bar{y}_s(k) > y_o(k)$,

and

$$\bar{y}_{ls}(k) = y_{ls}(k - 1),$$

if $\bar{y}_s(k) \leq y_o(k)$,

and

$$\bar{y}_{ls}(k) = (1 - a_l)\,|y(k)| + a_l\,\bar{y}_{ls}(k - 1),$$

if $\bar{y}_s(k) > y_o(k)$,

whereby the subscript l represents the long-time term, the subscript ln the long-time term of the noise, and the subscript ls that of the voice, a_l the specific coefficients of the time constant, which are selected here much greater than those used for calculating the control signal $y_s(k)$. For this reason, the relationship between the control signal $y_s(k)$ and the threshold value y_o is a simple criterion for voice recognition and for calculating the position of the threshold value y_o. This occurs according to the formula:

$$y_o(k+1) = w_n(k)\,\bar{y}_{ln} + (k)\,w_s(\bar{k})\,\bar{y}_{ls}(k),$$

whereby $w_n(k)$ and $w_s(k)$ are the specific weighting of the long-time noise and the long-time voice terms.

Proper hands-free communication, e.g., with a telephone, is possible with these algorithms, because the voice quality and the stability of the system can be improved in a simple way and the sound level can be increased, without the amplification having to be controlled manually. The algorithms are also suitable for adapting the environmental parameters to a system change. They can be also be used together with other algorithms to eliminate echoes instead of damping of the same.

Both during use of an analog system and in digital system, the compander characteristic curve and the receiving volume can therefore be always automatically adjusted with the invention depending on the speaking volume and ambient noises and thereby adapted to the acoustic situation. The receiving volume is thereby also raised at higher noise levels. With increasing control voltage U6, the maximum sensitivity is reduced by the characteristic curve displacement (displacement of p3 or y_o to the right). Simultaneously, higher noise levels as well are diminished furthermore through the expander range, which is displaced concurrently, whereas the reproduction volume changes proportionally the opposite way, therefore raised, and the voice signals continue to be sent through the compression at a constant level. Furthermore, the swing of the expansion is reduced, if the interval between the voice level and noise level becomes smaller, as is shown by way of example based on characteristic curve 24.1 in Figs. 1, 2, and 6. The constant adjustment of the reproduction volume to the requirement is maintained thereby.

Claims

1. Method for the improvement of the transmission properties of an electroacoustic device, which is provided with adaptive dynamic control and the transmitting branch of which has at least one microphone and a transmitting amplifier and a receiving branch with a receiving amplifier and at least one loudspeaker, as well as a compander for adaptive dynamic control, the starting point of the compression and expansion of which, depending on the transmitted, incoming, echo, and noise signals, can be changed in that these signals are determined and control signals are produced according to the level thereof and are used for compander control through characteristic curve change and/or displacement, particularly for improving the hands-free properties of a telecommunication terminal, **characterized in that**

   - the transmitted signal (U1 and/or U2) of the transmitting branch (2) is supplied to a voice detector (28), which recognizes in the voice frequency range the peak value (L_{ss}) of the transmitted signal (U1; U2), that a voice level corresponding to the peak level (L_{ss}) is determined, and a voice level control signal (USp) corresponding to said level and used to control the characteristic curve (LU2 = f(LU1)) is produced or calculated,
   - that the transmitted signal (U1; U2) is supplied simultaneously to a noise detector (29) and the noise level is determined and a noise level control signal (UG) corresponding to said noise level and used to control the characteristic curve (LU2 = f(LU1)) is produced or calculated, and

- that by means of addition of the voice level control signal (USp) and the noise level control signal (UG), the position of the characteristic curve (LU2 = f(LU1)) is established so that the determined voice level lies within the compression range (p2-p3) and the determined noise level below the expansion range (p1-p2).

2. Method according to claim 1, characterized in that the incoming signal (U10) of the receiving branch (22), particularly downstream of the volume control (17), is supplied to a threshold detector (36), the receiving level (|U12|) is determined, and a receiving level control signal (UE) is generated, if the receiving level (|U12|) is same as or greater than the threshold value (x), that with the receiving level control signal (UE) a control unit (35) is controlled, by which the compander control is interrupted and the last present level, effecting the compander control, of the control signal (U6) is stored and the compander control for the duration of the receiving level control signal (UE) is carried out as a function of the stored control signal (U6), and that if the receiving level control signal (UE) is not present, the compander control is again connected.

3. Method according to claim 1, characterized in that with an increasing noise level and/or with a declining interval between the voice level and noise level the swing of the expansion range (26) is reduced and reversed, in that either with a change in the determined noise level and/or the determined interval between the voice and noise level a control signal controlling the expansion swing of the compander (5) is produced and the expansion branch of the compander is switched on.

4. Method according to claims 1 and 2, characterized in that the components, determining the characteristic curve (LU2 = f(LU1)), of the compander (5) are selected so that first the basic adjustment of the characteristic curve occurs according to an average acoustic environment, that next it is determined whether an incoming signal (U12) is present, which is greater than the preset receiving level threshold value (x), that in the presence of an incoming signal greater than the threshold value (x) an anti-sidetone evaluation and by means of this a characteristic curve control occur, and if the incoming signal (U12) is not present or too small, it is determined whether a burst occurs in the transmitted signal (U1; U2), that if a burst is present, the level of the peak value (Lss) is determined, multiplied by an initial weighting factor (k1), and thus the position of the level (Up2; Up2-p3), determining the starting point (p2) or the range (p2-p3) of the compression, of the characteristic curve is determined or calculated, and if a burst is not present, the noise level is determined, and it is determined whether it is greater or less than the level of the starting point (p1) of the expansion, that at a smaller level the noise level determination is repeated and at a greater level said level is multiplied by a second weighting factor (k2) and thus the position of level (Up1), determining starting point (p1) of the expansion, is determined or calculated, that then from the sum of the weighted levels (Up1 and Up2 or Up1 and Up2-p3) the control signal (U6) used to control the characteristic curve is generated or calculated and the characteristic curve displacement is carried out with this.

5. Method according to claim 4, characterized in that to produce the time constants of the voice detector (28) and the noise detector (29) in each case an integrating element (48, 49) comprising a capacitor (C), resistor (R_x) with a diode (D) connected in series with said resistor and a resistor (R_y) is used, whereby the following conditions are met for the time function F(t):

6. Method according to claim 4, characterized in that to determine or calculate the position of the starting point (p2) or the range (p2 - p3) of the compression [and] the time constants of the voice filter (40) and the noise filter (42), an averaged value is generated or calculated in each case from the instantaneous value of the control signal (U6 . 1/(a + 1) and the previously determined value (U6 * . a) of the same and made available for compander control, whereby to calculate the time constants of the voice filter (40) and the noise filter (42) in each case values inversely proportional to each other are used for the factor a.

7. Method according to claim 1, characterized in that the control signal (y_s) to be emitted by the voice detector (28) is calculated from the transmitted signal (y(k)) for rising levels according to the formula

$$\bar{y}_s(k) = (1 - a_r) \mid y(k) \mid + a_r \bar{y}_s(k-1),$$

i.e., for the case that $|y(k)| > y_s(k-1)$, and for falling levels according to the formula

$$\bar{y}_s(k) = (1 - a_f)\,|\,y(k)\,| + a_f\bar{y}_s(k-1),$$

i.e., for the case that $|y(k)| \leq y_s(k-1)$, whereby a_i are the parameters for the time constants of the voice filter and a_r establishes the time constants for rising levels and a_f those for falling levels, and the time constants a_r and a_f are selected so that the rise time of the ramp slope is 5 to 20 times, especially 10 times shorter than that of the falling slope and that the amplification factor $(g(y_s))$ of the compander (5) is calculated with the thus obtained control signal $(y_s(k))$ according to the following formula:

$$g(\bar{y}_s) = C_1,$$

if $\bar{y}_s \leq \omega y_o$,

$$g(\bar{y}_s) = C_2\,\bar{y}_s^{\,P_r-1},$$

if $\omega y_o < \bar{y}_s \leq y_o$, and

$$g(\bar{y}_s) = C_3\,\bar{y}_s,$$

if $y_o < \bar{y}_s$.

8. Method according to claim 7, characterized in that the threshold value (yo), therefore the starting point (p2) of the compression is calculated according to the formula

$$y_o(k+1) = w_n(k)\,y_{ln}(k) + w_s(k)\,y_{ls}(k),$$

where $w_n(k)$ and $w_s(k)$ stand for weighting factors and the subscript n = noise portion, the subscript s = voice portion, the subscript ln = long-time term of the noise signal, and the subscript ls = long-time term of the voice signal and the last term is calculated according to the following formulas:

$$\bar{y}_{ln}(k) = (1 - a_l)\,|\,y(k)\,| + a_l\,\bar{y}_{ln}(k-1),$$

if $\bar{y}_s(k) \leq y_o(k)$,

and

$$\bar{y}_{ln}(k) = +\,y_{ln}(k-1),$$

if $\bar{y}_s(k) > y_o(k)$, and

$$\bar{y}_{ls}(k) = \bar{y}_{ls}(k-1),$$

if $\bar{y}_s(k) \leq y_o(k)$,

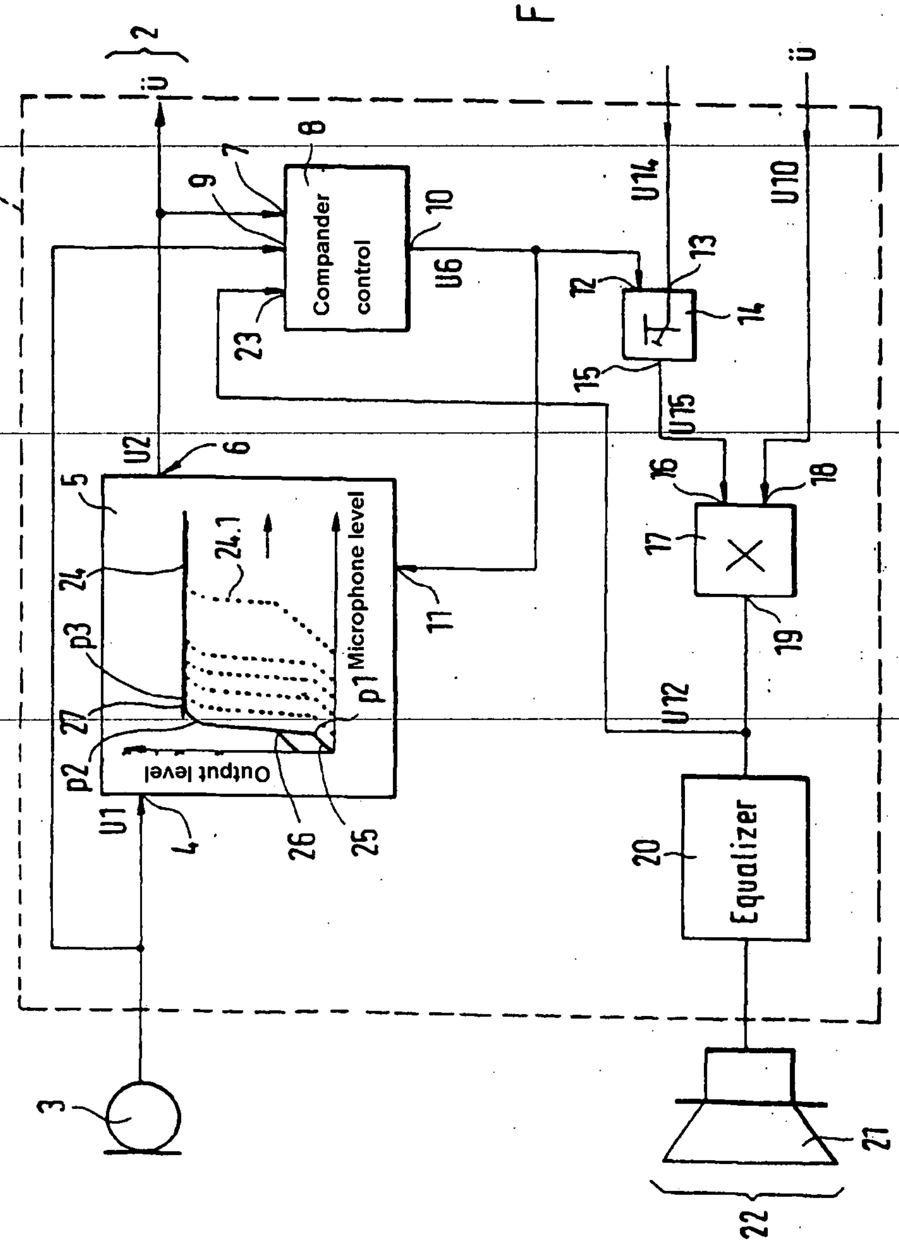
and

$$\bar{y}_{ls}(k) = (1 - a_l)\,|\,y(k)\,| + a_l\bar{y}_{ls}(k-1),$$

if $\bar{y}_s(k) > y_o(k)$.

9. Method according to any one of claims 1 through 8, characterized in that the compander is realized through a digital signal processor, that the analog transmitting and incoming signals (U1 and U10 or U12) are digitalized, and that the characteristic curve course is stored in digital form in a memory and is displaced and/or changed according to the digital control signals calculated from the individual control values, and the thus modified characteristic curve is used for compander control.

FIG.1


Compander control
Microphone level
Output level
Equalizer
U1
U2
U6
U10
U12
U14
U15
p1
p2
p3
24.1
ü
ü

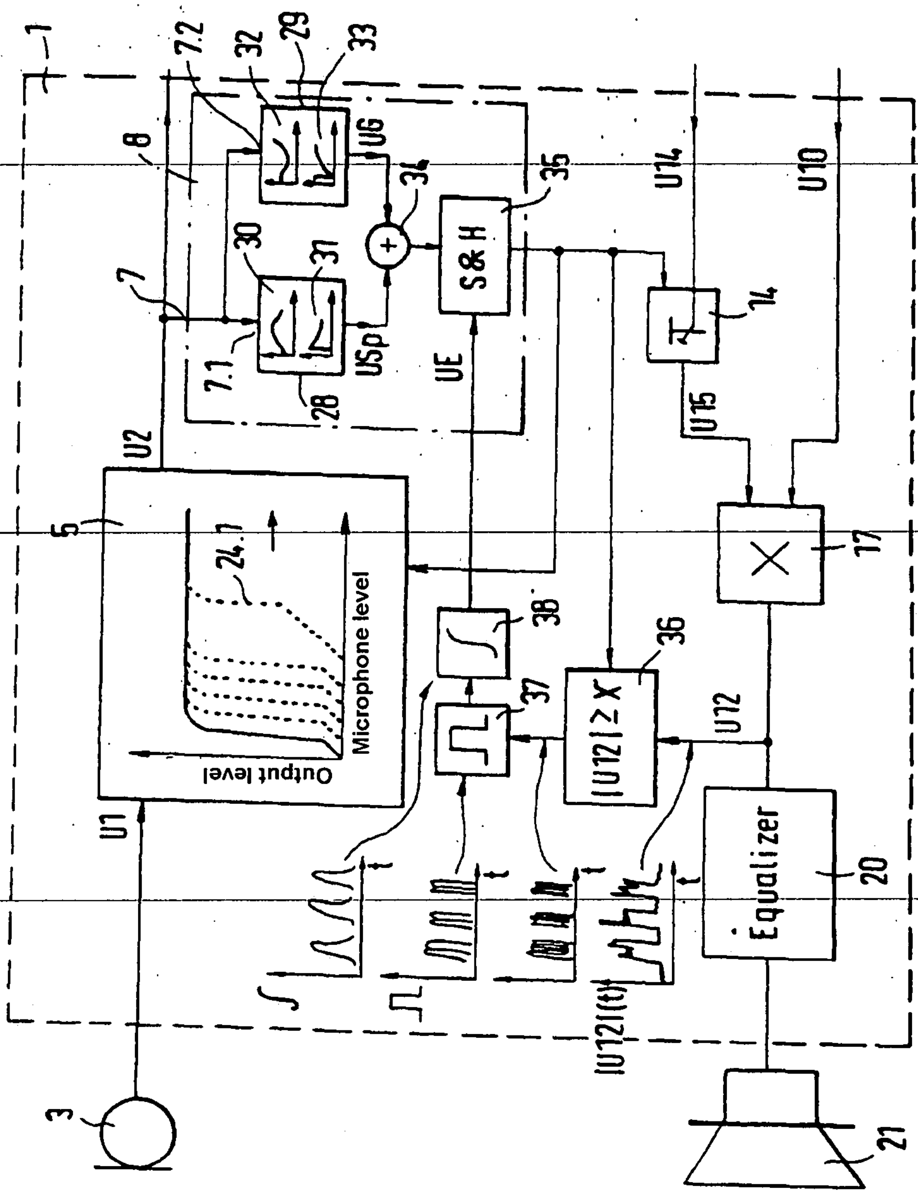

1
7.2
32
29
33
8
UG
34
35
30
31
7
S&H
U14
U10
14
7.1
28
USp
UE
U15
U2
5
24.1
Microphone level
Output level
38
37
36
17
|U12| ≥ X
U12
U1
Equalizer
20
|U12|(t)
3
21

FIG. 2

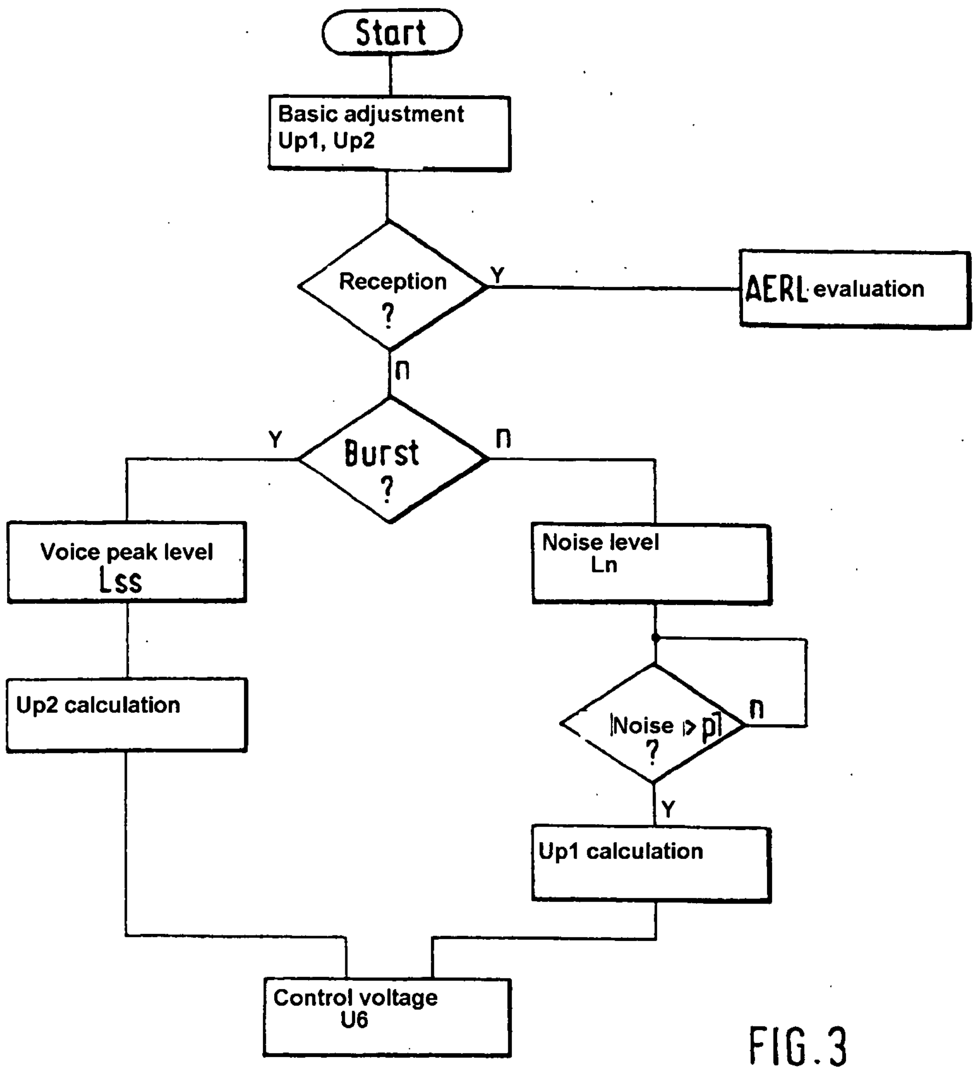

Start
Basic adjustment
Up1, Up2
Reception
?
Y
AERL evaluation
n
Burst
?
Y
n
Voice peak level
Lss
Noise level
Ln
Up2 calculation
|Noise |> p1
?
n
Y
Up1 calculation
Control voltage
U6

FIG.3

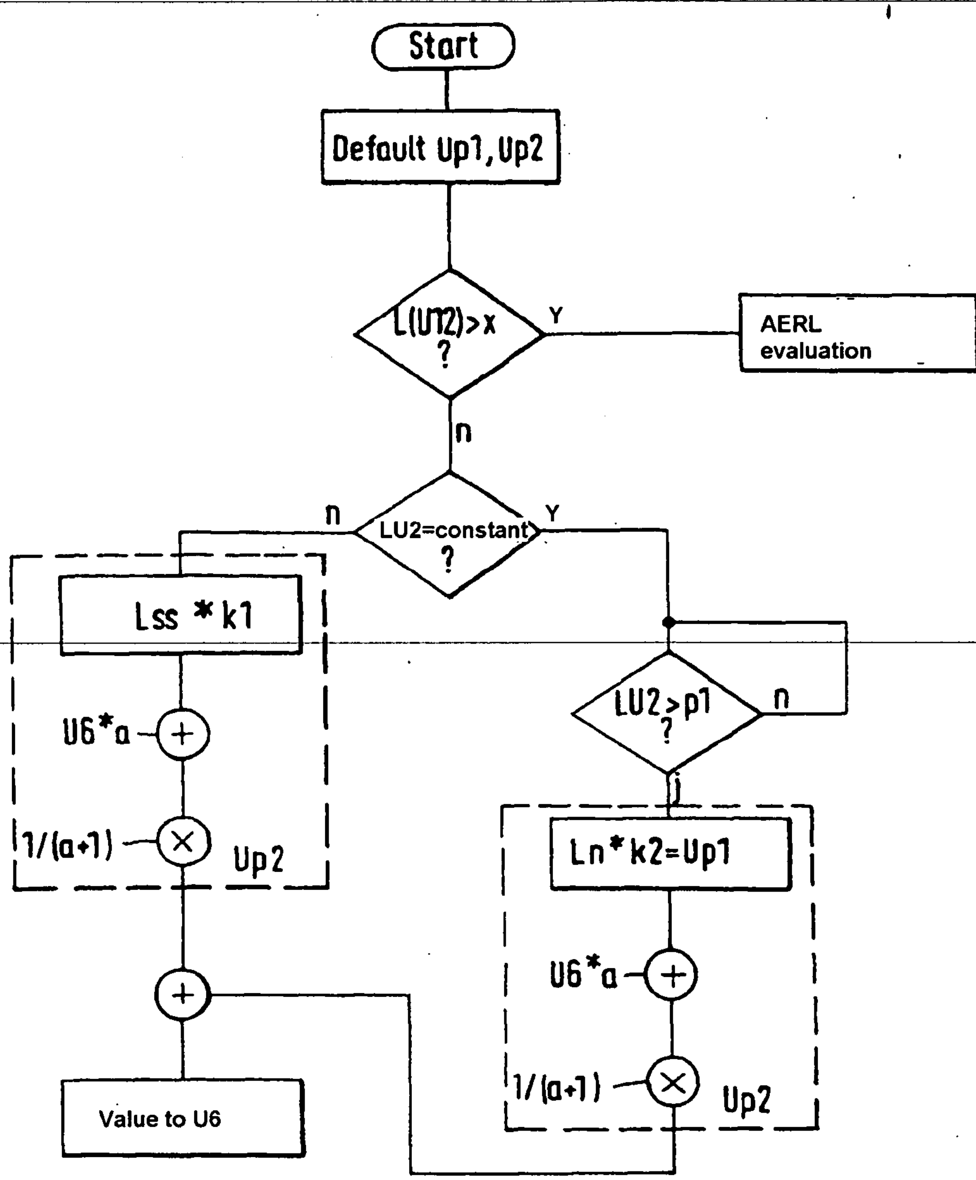

Start
Default Up1, Up2
L(UT2)>x ?
Y
AERL evaluation
n
LU2=constant ?
n
Y
Lss * k1
U6*a
1/(a+1)
Up2
LU2>p1 ?
n
j
Ln*k2=Up1
U6*a
1/(a+1)
Up2
Value to U6

FIG.4

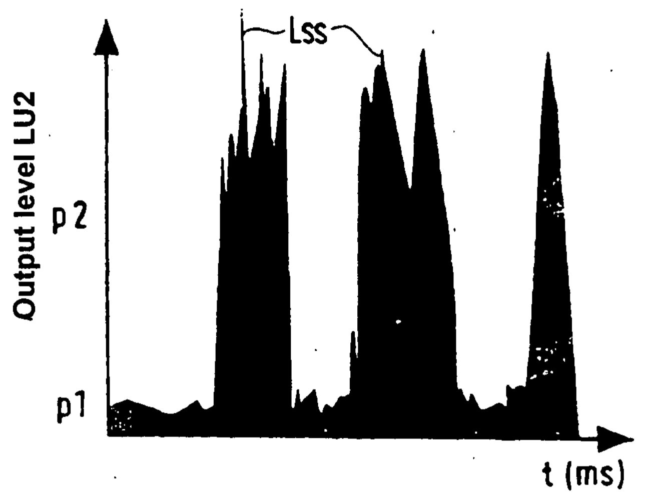

Lss
Output level LU2
p2
p1
t (ms)

FIG.5

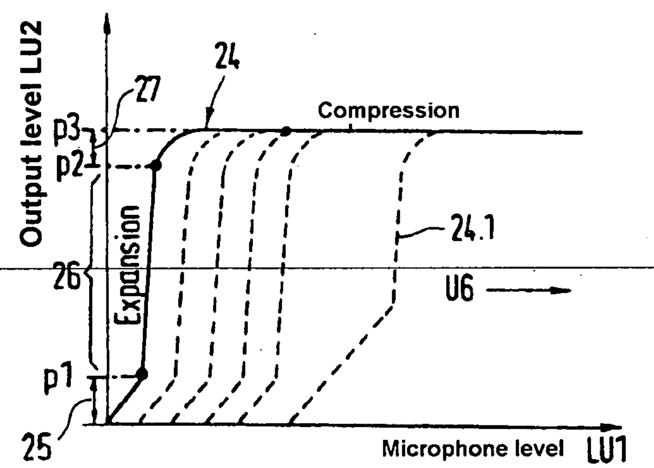
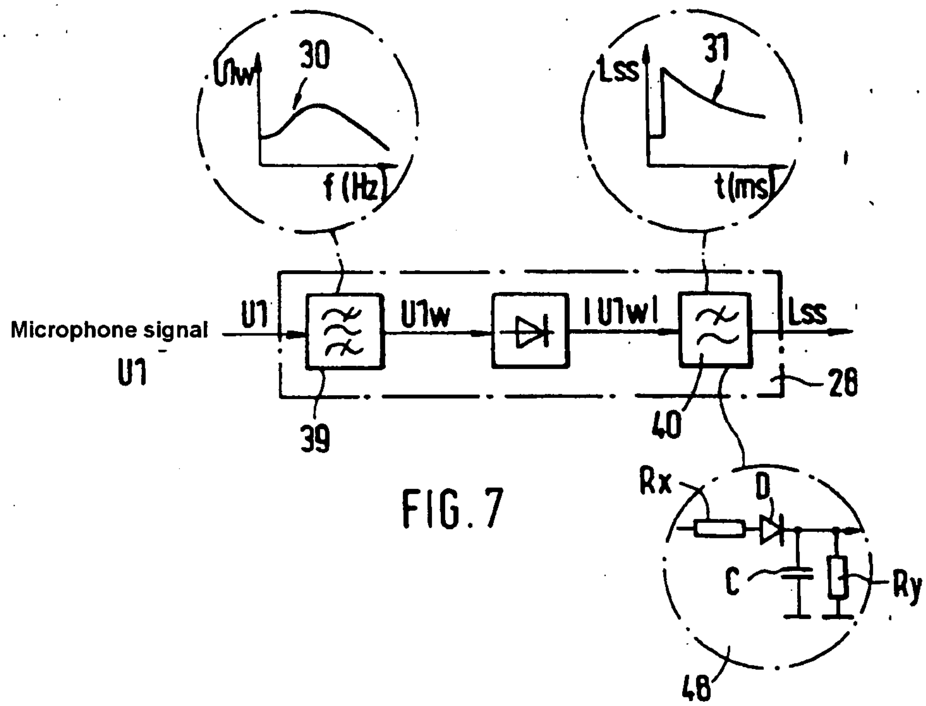
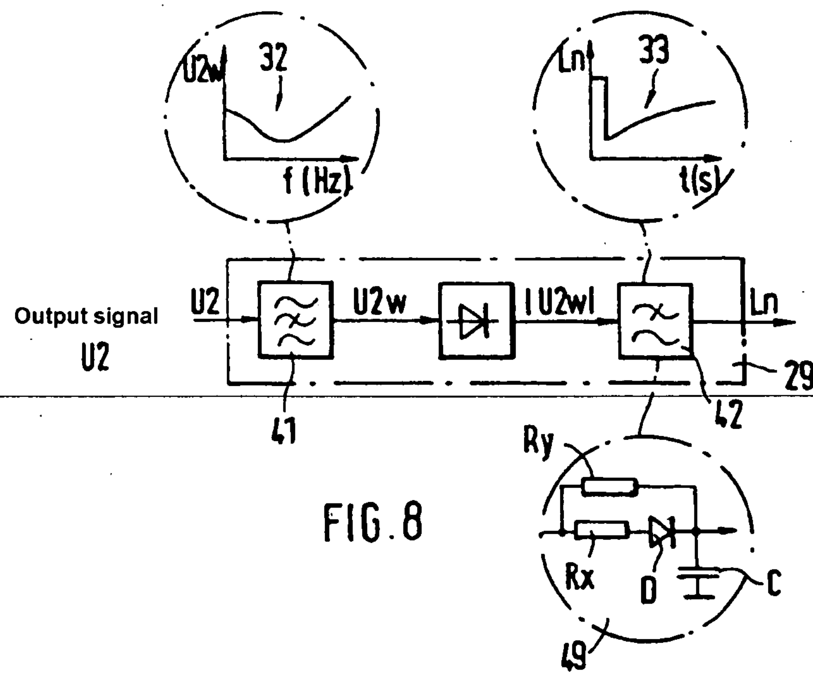

Output level LU2
24
27
Compression
p3
p2
Expansion
24.1
26
U6
p1
25
Microphone level
LU1

FIG.6



U1w
30
f (Hz)
Lss
31
t(ms)
Microphone signal
U1
U1
U1w
|U1w|
Lss
28
39
40
Rx
D
C
Ry
48

FIG. 7



U2w
32
f (Hz)
Ln
33
t(s)
Output signal
U2
U2
U2w
|U2w|
Ln
29
41
42
Ry
Rx
D
C
49

FIG. 8

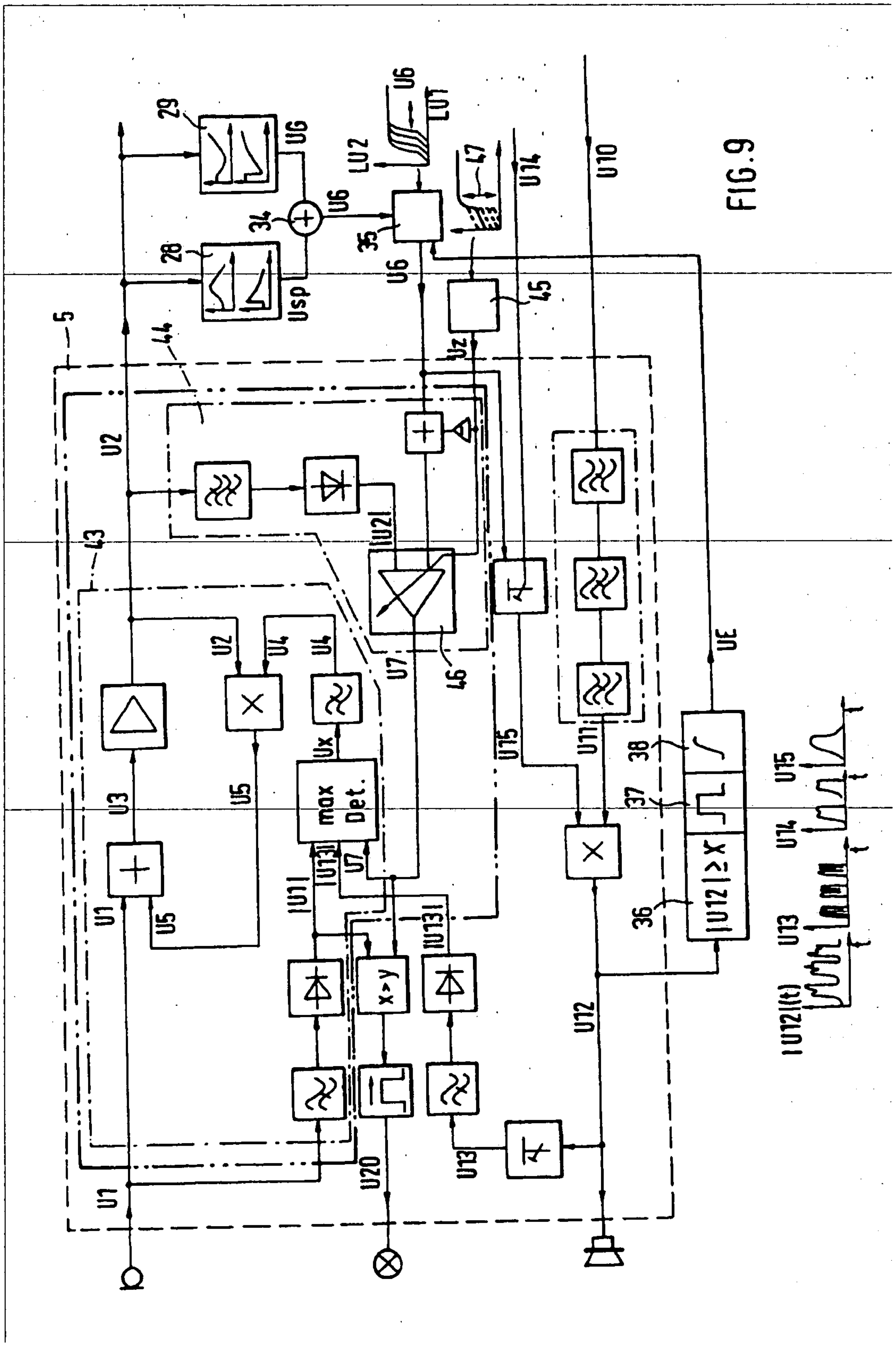

U1
U2
U3
U5
U6
UG
Usp
U7
Ux
Uz
UE
U10
U11
U12
U13
U14
U15
U20
max Det.
x>y
LU2
LU7
5
28
29
34
35
36
37
38
43
44
45
46
47

FIG.9

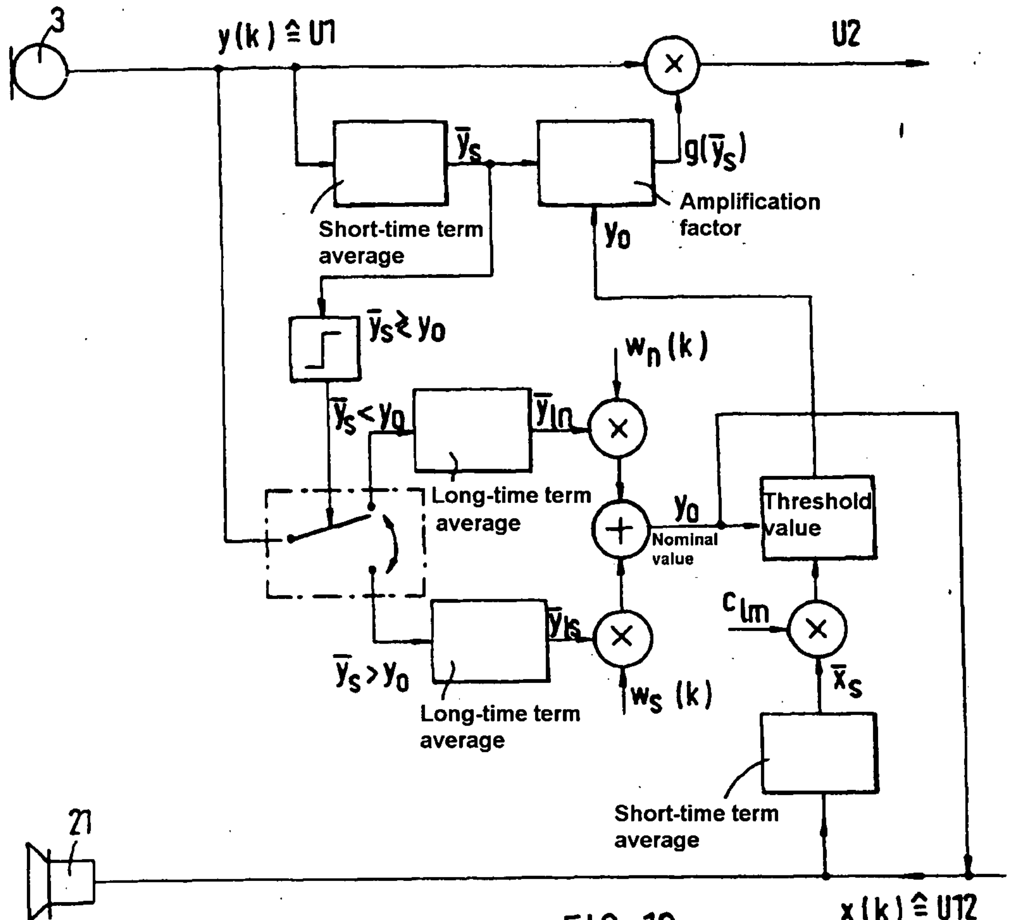

3
y(k) ≙ U1
U2
$\bar{y}_s$
g($\bar{y}_s$)
Short-time term average
Amplification factor
y_0
$\bar{y}_s \gtrless y_0$
$w_n(k)$
$\bar{y}_s < y_0$
$\bar{y}_{ln}$
Long-time term average
y_0
Nominal value
Threshold value
$\bar{y}_{ls}$
c_{lm}
$\bar{y}_s > y_0$
$w_s(k)$
$\bar{x}_s$
Long-time term average
Short-time term average
21
x(k) ≙ U12

FIG. 10

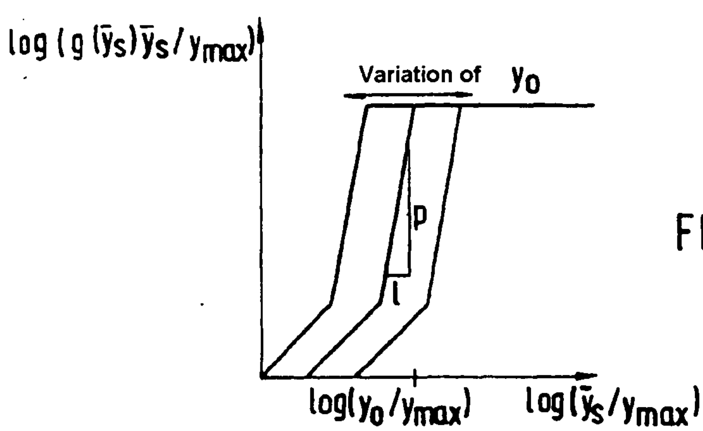

log (g($\bar{y}_s$)$\bar{y}_s$/y_{max})
Variation of y_0
p
l
log(y_0/y_{max})
log($\bar{y}_s$/y_{max})

FIG. 11

European Patent Office

EUROPEAN SEARCH REPORT

Application Number

DOCUMENTS CONSIDERED TO BE RELEVANT			EP 93113875.4
Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim	CLASSIFICATION OF APPLICATION (Int. Cl.5)
A	DE - A - 3 822 353 (SIEMENS) * All * --	1	H 04 R 3/02
A	EP – A – 0 467 99 (PIONEER) * Abstract; column 1, line 1 – column 2, line 8; Fig. 4, Claim 1 * --	1	
A	EP – A – 0 280 907 (SIEMENS) * Abstract; column 1, line 1 – column 2, line 32; Fig. 1; claim 1 * -----	1	
			FIELDS SEARCHED (Int. Cl.5)
			H 04 R

The present research report was prepared for all patent claims

Research site	Date of completion of research	Examiner
Vienna	10-11-1993	GRÖSSING

CATEGORY OF CITED DOCUMENTS
X: of particular relevance considered alone
Y: of particular relevance in association with another document in the same category
A: technical background
O: oral disclosure
P: literature published in interim
T: theories or principles underlying the invention

E: earlier patent document but published on or after the filing date
D: document cited in application
L: document cited for other reasons

&: member of same patent family, conforming document